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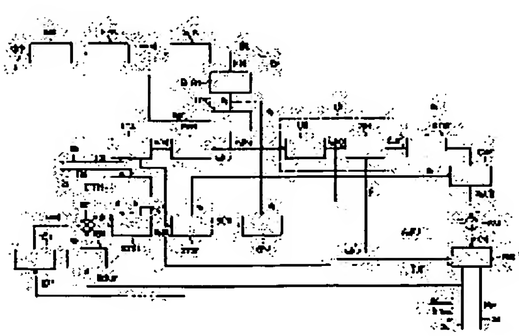
(54) VOICE ENCODER ADOPTING ANALYSYS-SYNTHESIS TECHNIQUE BASED ON PULSE EXCITATION



(57)Abstract:

PURPOSE: To use a voice encoder, where an analysis-syntnesis technique is adopted, with a satisfactory performance.

CONSTITUTION: In an analysis-synthesis encoder, an original voice signal is shifted by a short time so that it is consistent with an expected signal, which is encoded with a replica made by a long-time synthesis filter, with respect to time. This shift is determined in each subframe by through search in the range of such possible values that energy of an error signal is minimized. Once the optimum shift is determined, an optimum excitation is searched. This excitation has a very small number of pulses arranged in a decision theoretic constitution and is selected in a code book including words all of which are obtained from a limited

number of keywords. The decision theoretic constitution realizes quick search of the optimum excitation without storing the code book neither actually executing the synthesizing filter operation of candidate excitations.

CLAIMS

[Claim(s)]

[Claim 1] a coding step -- the next actuation, i.e., :-, several [sampled the original sound signal in the 1st rate of a sampling and the 1st was beforehand decided to be for each in the sequence of a sample [x (n)] produced as a result] -- the sample of Ls -- or the sample which two or more blocks containing the integral multiple of said 1st number follow -- dividing --;

- In order to decide one group's linearity preliminary announcement multiplier (ai) used to linearity preliminary announcement ********, short-term composition ********, and spectrum weighting ********, short-term analysis is performed about the original sound signal. The display of said multiplier in the frequency domain is generated, and it inserts in the signal which it related [signal] to the value of said display and had effective information [j (phi)] encoded over a period equal to the period of the sample of the block with which 1 block or one group continues.;

- It lets said linearity preliminary announcement ******** pass, and the short-term residual signal [rs (n)] over the sample of said block or a block of a group is acquired.;

- In order to determine a solution parameter over a long period of time containing the long-term composition **** delay d and a multiplier b, long-term analysis is perform about said residual signal [rs (n)], and it inserts in the signal which had effective information [j (d), j (b)] encode over time amount equal to the period of the sample of the block with which 1 block or one group continues in relation to the value of said parameter.;

- Each consists of an amplitude contribution (excitation gain) and a gestalt contribution (renovation). The latter consists of a limited number far smaller than the sample of said 1st number of pulses. The location and amplitude which were specified beforehand belong to the set of each finite. About the excitation signal chosen within the one-set excitation signal, long-term composition ********, The sound signal sample of each block of the schedule encoded as it is also with the sound signal [yw (n)] which is acquired by performing short-term composition ******** and spectrum weighting ********, and by which weighting was reconfigurated and carried out is reproduced.;

- The time amount shift of the one-set sample of said residual signal [rs (n)] is carried out by the discrete step, and it sets there. The residual signal sample of each set In order to align the reconfigurated residual signal [ss (n)] which is acquired as a result of short-term composition ******** about an excitation signal, and its residual signal in time amount It has a number equal to the measurement size in one block of the sound signal sample which should be encoded of samples. correction sound signal [xw by which the shift there generated the corrected residual signal [rm (n)] which is exposed to the same short-term composition ******** as what was performed to said

excitation signal, and spectrum weighting ********, was reconfigurated by this, and weighting was carried out -- (n) --] -- making --;

- It is reconfigurated with the correcting signal [xw (n)] by which weighting was reconfigurated and carried out. The optimal excitation signal over the sample of each block is determined by making into min energy of the error signal [e (n)] by which weighting was carried out expressed according to the difference between the signals [yw (n)] by which weighting was carried out. And actuation inserted in the signal which had the information (j (s), and [j (gmax), j (gnor), sigma]) which identifies the optimal excitation signal encoded is included. A sound signal is set to the approach of encoding / decrypting, and it is :-. Said renovation pulse is only a sample which is not the zero of the word which consisted of samples of said 1st number Ls. - The word of the group to whom, as for the renovation word over the excitation signal of the 1st subset, the 1st set was limited including one pair of pulses Two pulses are the keywords put on the key position decided beforehand. Other words in the subset those pulses -- 1 time -- one location -- the edge of a word -- going -- one of the pulses of these -- said edge -- or until it arrives at the key position of other pulses in the start word By shifting to coincidence, it is obtained from each of the keyword, the direction of the shift is the same to all words, and they are; and -. The renovation word over the excitation signal of the 2nd subset Include only one pulse from which the location differs to each signal, and said decision of; and the optimal excitation signal is received. It is direct calculated by using pulse response [of the filter with which the energy of said error signal by which weighting was carried out performs the composition about the excitation signal, and spectrum weighting ********] Q (n). In the count, it is the next actuation, i.e., :-. Said pulse response Q (n) to each of the possible pulse position in an excitation signal and its energy Eq are determined.;

- The energy of the 1st partial error signal [e1 (n)] and same error signal expressed with the difference between the contributions [yw 1 (n)] of a signal [xw (n)] and excitation signal **** memory by which weighting was reconfigurated and carried out is determined.;

- The 1st correlation R (e1 q) between pulse response Q (n) to each of said 1st partial error signal [e1 (n)] and the pulse of an excitation signal is determined.;

- To each excitation signal, said pulse response is left and the signal [u (n)] showing the contribution of ******** by the initial condition of the zero of the excitation signal is determined.;

- The 2nd correlation R (e1 u) between said signal [u (n)] and the 1st partial error signal [e1 (n)] which express the contribution of ******** by the initial condition of the zero of an excitation signal to energy [of said signal [u (n)]] E (u) showing the contribution of ******** by the initial condition of the zero of an excitation signal and a list is determined.;

- The optimum value of the amplitude contribution is determined to each excitation signal as a ratio between said 2nd correlation and the energy of a signal produced from ******** in the initial condition of zero.;

- Approach characterized by including the actuation which calculates the value of the error signal energy over each excitation signal as a function of said 2nd correlation R of said energy E of said energy Eu of the signal showing the contribution of ******** by the initial condition of the zero of

excitation and its 1st partial error signal (e1) (e1 u).

[Claim 2] said pulse -- single -- the approach of claim 1 characterized by having the amplitude of 1.

[Claim 3] The sequence of a sound signal sample is constituted by the reverse frame which the plurality to which each supports one of said the blocks follows. And it is divided to the frame containing a number Lf of samples with which the 2nd was decided beforehand, said short-term analysis is performed to each frame, and said short-term analysis in a frame is received. The die length is Lf+P (the number of the linearity preliminary announcement multipliers in P= each group). The sample aperture also containing the sample of several H+K with which included a present frame and the continuing present frame, and said continuing frame was beforehand decided to be is analyzed. Said aperture It is the approach of claim 1 characterized by being the trapezoidal shape aperture which sets aside the 1st and the last P sample, and carries out weighting of all the samples to it being also at the greatest weight, and determining the weighting factor to it through the linear interpolation between the minimum weight and the greatest weight.

[Claim 4] The linearity preliminary announcement multiplier ai is the approach of claim 3 which is the multiplier obtained as a result of interpolation between the value given in the short-term analysis to the present frame, and the value given to a front frame, and is characterized by performing interpolation there by operating about said display to the reverse frame in early stages of each frame.

[Claim 5] Said linearity preliminary announcement residual signal is the approach of any one publication in the above-mentioned claim which receives ******** of reduction in advance of long-term analysis, and is characterized by giving the residual signal [rf (n)] ****(ed) by that cause.

[Claim 6] The sequence of a sound signal sample is divided to the frame containing a number Lf of samples with which each consisted of two or more reverse frames corresponding to one of said blocks to follow, and the 2nd was decided beforehand. In order to perform said long-term analysis to each frame and to determine a solution parameter over a long period of time in the first half present -- a frame -- continuing -- a frame -- including -- and -- said -- continuing -- a frame -- beforehand -- specifying -- having had -- several -- H -- + -- K -- a sample -- containing -- **** -- ****(ing) -- having had -- the remainder -- a signal -- [-- rf -- (-- n --) --] -- a sample -- an aperture -- analyzing -- having -- things -- the description -- ** -- carrying out -- a claim -- one -- an approach .

[Claim 7] It be the approach of claim 6 characterize by determine the gain in each frame including the actuation as which said long-term analysis be determine in the long-term preliminary announcement gain G showing the ratio between the energy of the ****(ed) residual signal in the input of a means to perform said analysis to each term further , and the output from this means .

[Claim 8] said long-term analysis -- further -- the next actuation, i.e., :-, uttering the sound signal corresponding to a frame depending on the value of said long-term analysis multiplier b, and the preliminary announcement gain G -- or -- the case where classified and it is classified as whether it is uttered as that by which the segment is uttered -- the 1st flag (V) -- generating --;

- The multiplier b relevant to the value and the present frame of the long-term analysis delay d is compared with the thing relevant to a front frame. Fewer than the amount as which delay

fluctuation was specified beforehand, when the multiplier value in both frames is forward The approach of claims 6 or 7 characterized by including the actuation which generates the 2nd flag (F) which enables interpolation between the delay and the multiplier value which were calculated to the frame of said front, and the thing calculated to the present frame.

[Claim 9] The long-term analysis delay d is determined as max of the automatic correlation function of the ****(ed) residual signal within the aperture used to the analysis itself. Before determining the preliminary announcement gain G over the long-term analysis multiplier b and the present frame, and the maximum of said automatic correlation function If said 1st and 2nd flags are generated with the frame of said front, it will be determined even [near the maximum of the same function in a front frame]. And said maximum An approach given in any of claims 6-8 which will be characterized by being used as delay to the present frame if only amounts smaller than the value as which it was beforehand specified from the max in the aperture relevant to the present frame differ they are.

[Claim 10] The value of the long-term analysis multiplier b be the 1st maximum b1 coordinated with the ratio between the energy of the ****(ed) residual signal in the frame before [in the present frame] it reach and the die length be in spacing equal to the long-term analysis delay . Approach given in any of claims 6-9 characterize by clip they be .

[Claim 11] For the value of the long-term analysis multiplier b, while the preliminary announcement gain G is below the gain threshold Gthr it is the 2nd maximum b2. If it exceeds, it will be the 2nd maximum b2. Approach given in any of claims 6-10 characterized by clipping they are.

[Claim 12] the long-term analysis delay d and said interpolation of a multiplier b be the residual signal ss which be the linear interpolation which extend over all frames, and be reconfigurated in the case of the interpolate [nonintegral] delay value. it be the approach of claim 8 characterize by estimate that the secondary polynomial interpolation by which the core be established in the perimeter of the integer delay value nearest to said interpolate value be also for the value of the sample to which (n) correspond.

[Claim 13] The information relevant to the long-term analysis multiplier b inserted in the encoded signal is an index showing the quantized multiplier value. And a multiplier value smaller than the fraction as which the value by which the information relevant to the long-term analysis delay d expressed the delay value on the outside of spacing of the permitted delay, and min was quantized was specified beforehand is set to 0. An approach given in any of claims 6-12 characterized by being inserted in the signal with which the index showing the delay information showing the value on the outside of said spacing of the permitted delay and the value by which said min was quantized was encoded when made 0 they are.

[Claim 14] in order to determine the optimal excitation, as for the excitation signal of said 2nd subset, said 1st flag (V) was generated -- or the approach of claims 1 and 8 which will be characterized by the thing the analysis of the energy distribution in the residual signal which was used according to whether said flag was generated and was corrected indicates energy

concentration in a short time to be, and for which it shows the standup of an utterance sound if it becomes.

[Claim 15] It is the approach of claim 14 characterized by normalizing the excitation signal of two subsets as it is also at a different normalization factor coordinated with the number of the pulses in each subset signal in order to determine the optimal excitation.

[Claim 16] It is said 1st flag (V). An amplitude contribution of as opposed to [if generated] the excitation signal of said 2nd subset is the approach of claim 14 characterized by being restricted so that the threshold proportional to the absolute value of the residual signal may not be exceeded.

[Claim 17] Said analysis of the energy distribution of said corrected residual signal is performed in each reverse frame. (And the next actuation, i.e., :-, Said reverse frame is divided into two or more apertures which overlapped partially, and the aperture of the beginning and the last supports each first stage or last section of the reverse frame. There) the aperture following the first thing shifts only one sample about each and the aperture before that -- having -- coming -- **** --; -- further - the energy of the corrected residual signal in all reverse frames, power, and the energy in one each of said the apertures -- determining --;

- The power to the aperture the energy of whose is max is determined, the ratio between the power in said aperture and the power in a reverse frame is determined, and they are; and -. The approach of claim 14 characterized by including the actuation said energy concentration will be recognized to be in said maximum energy and said power ratio if said maximum energy and said ratio are below each threshold there as compared with each threshold.

[Claim 18] An approach given in any of claims 6-17 characterized by being restricted to the maximum as which only the amount to which the long-term analysis delay d is proportional to the whole shift accumulated by the front frame was changed, and the **** value of the fluctuation was specified beforehand if the 2nd flag (F) is generated they are.

[Claim 19] Thereby, fluctuation of said delay is the approach of claim 18 which changes the decision about interpolation and is characterized by being made an invalid if made as [come / out of the value of spacing which was able to opt for delay beforehand].

[Claim 20] If at least one of said the 1st and 2nd flags is generated, said residual signal If the analysis of the corrected residual signal energy in a reverse frame shows that the sound signal segment which receives said time amount shift in a reverse frame, and corresponds is not silence, and includes the pitch peak All shifts in a frame since the shift relevant to a reverse frame is accumulated with the thing of the reverse frame in front of the same frame are approaches given in any of claim 1 characterized by remaining below in the maximum shift, and claims 6-19 they are.

[Claim 21] Said analysis of the corrected residual signal energy is the next actuation, i.e., :-. When it reaches in the energy itself, it compares with the energy threshold which shows that a sound signal segment is not silence.;

- The corrected residual [in / it reaches and / spacing with the die length equal to long-term analysis delay] signal power in a reverse frame and the ratio between these power are determined, and they are; and -. The approach of claim 20 characterized by include the actuation in comparison with the

power threshold which shows existence of the pitch peak in a reverse frame when this ratio is exceeded.

[Claim 22] It is the approach of claims 20 or 21 which the shift to a reverse frame is determined within spacing which extends around the shift accumulated by the reverse frame in front of the same frame before determining the optimal excitation signal, and are characterized by it being a value which makes min energy of said first partial error signal [e1 (n)].

[Claim 23] A shift [in / in order to determine said shift, the rise sampling of said residual signal is performed in the 2nd rate which is the multiple of the 1st rate, and / a reverse frame] is the approach of claim 20 characterized by being equal to the sample beyond one or it of a residual signal by which the rise sampling was carried out.

[Claim 24] The signal showing the correction residual signal ****(ed) as said 1st partial error signal is also at the initial condition of zero [xw2 (n)], And it is calculated as the sum between the 2nd partial error signal [e0 (n)] which is a difference between the memory contribution [xw1 (n)] of correction residual signal ********, and the memory contribution [yw1 (n)] of excitation ********. The signal [xw2 (n)] showing the correction residual signal ****(ed) as it is also at the initial condition of the zero relevant to the sample in a reverse frame As opposed to each of the remaining shift [in / it is obtained by performing actual **** of the corrected residual signal to the shift value between the upper limit of the spacing, and the mean value between two extremal value, and / another side and its spacing] It is the approach of claims 22 or 23 characterized by being repeated from said pulse response from the value relevant to a front sample.

[Claim 25] The decision of said spacing of a shift value is the next actuation, i.e., :-. Two symmetry values about the accumulated value are defined to the spacing edge.;

- The residual signal peak location in the residual signal by which the rise sampling was carried out is determined, and it is compared with the peak location in a front reverse frame.;

- The escape of the spacing [be / by the consequent duplicate or loss of a residual signal peak / it / in order to avoid too much shift of the reverse frame in the past and/or the future] on one side of the accumulated value or both sides is restricted. The approach of claim 24 characterized by performing through actuation.

[Claim 26] In the spacing limit only in one side of the accumulated value, the search to the shift is the approach of claim 25 characterized by performing in consideration of a certain fixed number beyond the spacing edge which is unrelated to said limit so that equally to the number of values with which the number of the examined value on the whole is contained between said symmetry values.

[Claim 27] The information (j (phi), j (d), j (b), j (s), and [j (gnor), j (gmax), sigma]) about a solution parameter and an excitation signal is left a linear preliminary announcement multiplier display and over a long period of time. Said display is reconfigurated and the preliminary announcement multiplier of the reconfigurated linearity is obtained from it. A solution parameter is reconfigurated over a long period of time, and an excitation signal is chosen in the one-set excitation signal corresponding to what was used at the coding step. And since said signal generates the sound signal

sample [y (n)] of the reconfigurated block to each excitation signal [s (n)] By using the preliminary announcement multiplier ai of the reconfigurated linearity, the long-term analysis delay d, and a multiplier b Each block of the reconfigurated sound signal [y (n)] including the decryption step of receiving the same short period as what was performed at the coding step, and long-term composition ******** Preliminary announcement multiplier ai of the reconfigurated linearity which is obtained in the initial part of the shelf-life of a linearity preliminary announcement multiplier as a result of interpolation between the reconstruction value relevant to a very last shelf-life, and the reconstruction value relevant to the present period It is generated by performing short-term composition ******** to depend. If it is below the amount with which the value of the long-term analysis delay d and the value of a multiplier b relevant to two continuing shelf-lives were compared, and the delay fluctuation was beforehand decided to be and the multiplier is forward in both periods An approach given in any of claims 1-26 characterized by generating the flag corresponding to the 2nd flag, and enabling interpolation between solution parameter values in long-term composition ******** over a long period of time relevant to said two shelf-lives they are.

[Claim 28] It is equipment for using an analysis-composition technique, and encoding / decrypting a sound signal, and the decoder of a there is :-. Means (MT) of sampling a sound signal at the 1st rate, and dividing the sample sequence to the block which consists of a sample of the 1st number;

- One or linearity preliminary announcement multiplier ai of one group to the sample of a block beyond it It calculates. Said multiplier is changed into the display in the frequency domain. From said display The index j (phi) which identifies the multiplier itself which should be inserted in the encoded signal is obtained, and it has a short-term analysis means (STA, STR1) for leaving said index and reconfigurating a multiplier. There each group's linearity preliminary announcement multiplier -- one or the time amount period equal to the period of the sample of a block beyond it -- crossing -- effective --; -- further - The block of a signal sample is received from said sampling means (MT), and it is the linearity preliminary announcement multiplier ai. It receives from said short-term analysis means (STA, STR1), and is the short-term preliminary announcement residual signal rs. Linear preliminary announcement filter which generates (n) (LPC);

- The parameter for long-term composition ******** containing delay (d) and a multiplier (b) is obtained from said residual signal. It has a long-term analysis means (LTA, LTR1) for changing said parameter into the index [j (b), j (d)] of the schedule inserted in the encoded signal. And there a long-term solution parameter -- one or the time amount period equal to the period of the sample of a block beyond it -- crossing -- effective --; -- further - The long-term composition filter which receives said parameter from a long-term analysis means (LTA, LTR1) (LTS1), Said short-term analysis means (STA, STR1) to said linearity preliminary announcement multiplier ai The short-term composition filter (STS1) and spectrum weighting filter (SW) to receive are included with a series of configurations. The signal belonging to the one-set excitation signal with which each includes the gestalt contribution from which the number consisted of a number far smaller than said 1st number of pulses, the amplitude specified beforehand, and locations is received. And signal yw reconfigurated to one each of the excitation signal of the 1st **** system which generates (n) (LTS1,

STS1, SW);

- Reconfigurated residual signal ss which is generated by the long-term composition filter (LTS1) of said 1st **** system It is the one-set sample yw of said residual signal in order to make it align in (n) and time amount. (n) By the step to disperse, it has a means (TS) for carrying out a time amount shift. There The sample in the set of a residual signal has a number equal to the sample of said 1st number of samples. It is chosen within spacing of the permitted value and each shift step is; and also -. A series of the same short-term composition filters and spectrum weighting filters are included in the thing (STS1, SW) of said 1st **** system. The correction residual signal generated by the time amount shift means against each of the value of said spacing is supplied. It has the 2nd **** system (STS', SW') which generates the correction residual signal by which weighting was reconfigurated and carried out. Said 1st and 2nd **** systems (LTS1, STS1, SW1, STS', SW') the contribution showing the memory of front ********, and the contribution showing ******** by the initial condition of zero -- separate -- determining --; -- further - By generating the error signal [e (n)] by which weighting was carried out by comparing the signal generated by said 1st and 2nd **** systems, and minimizing the energy of said error signal by which weighting was carried out the means (SM, EM) for identifying the optimal excitation signal and the optimal shift, and inserting in the signal which had the information which identifies the optimal excitation signal encoded -- having --; -- and -- further -- a coding side -- :- Said index is left. Means for reconfigurating a solution parameter a linearity preliminary announcement multiplier and over a long period of time (LTR2, STR2);

- In the set corresponding to the set which is the coding side and was used for the thing (LTS1, STS1) of said 1st **** system including a series of the same long-term composition filters and short-term composition filters Let the information relevant to the optimal excitation pass, and the selected excitation signal is ****(ed). the equipment with which the 3rd **** system (LTS2, STS2) which generates the reconfigurated 1-block sound signal sample is formed -- setting -- :- a renovation pulse -- several [said / 1st] -- the sample which is not the zero of the word which consisted of samples of Ls -- it is --;

- The word of the group to whom, as for the renovation word over the excitation signal of the 1st subset, the 1st set was limited including one pair of pulses Two pulses are the keywords put on the key position decided beforehand, and other words in the subset the edge of a word -- going -- at once -- one location -- one of the pulses of these -- said edge -- or until it arrives at the key position of other pulses in the start word It is obtained from each of the keyword by shifting these pulses to coincidence. To all words, it is the same and the shift direction is; and -. The renovation word over the excitation signal of the 2nd subset Include only one pulse from which the location differs to each signal, and it sets for; and said error signal generating means (SM, EM). The means which makes error energy min consists of processing units, and this processing unit is :-. Said pulse response [Q (n)] to one each and its energy [Eq] of the possible pulse position in an excitation signal are determined.;

- The correcting signal [xw (n)] by which weighting was reconfigurated and carried out, the 1st

partial error signal [e1 (n)] expressed according to the difference between the contributions [yw1 (n)] of excitation signal **** memory, and the energy of the error signal itself are determined.;

- The 1st correlation [R (e1 q)] between the pulse responses to each of the pulse of said 1st partial error signal [e1 (n)] and excitation signal is determined.;

- To each excitation signal, said pulse response is left and the signal [u (n)] showing the contribution of ******** by the initial condition of the zero of the excitation signal is determined.;

- The 2nd correlation R (e1 u) between said signal [u (n)] and the 1st partial error signal [e1 (n)] showing the contribution of ******** of being also at the initial condition of the energy [E (u)] of said signal [u (n)] showing the contribution of ******** of being also at the initial condition of the zero of an excitation signal, and the zero of an excitation signal is determined.;

- It determines as a ratio between the energy of the signal produced from ******** of being also at said 2nd correlation and the initial condition of zero about the optimum value of an amplitude contribution to each excitation signal.;

- Equipment characterized by being arranged so that the error signal energy value over each excitation signal may be calculated as a function of said 2nd correlation R of said energy [E (e1)] of said energy (Eu) of the signal showing the contribution of ******** of being also at the initial condition of the zero of the excitation, and said 1st partial error signal (e1 u).

[Claim 29] Equipment of claim 28 characterized by preparing the low-pass filter (FPB) between said linearity preliminary announcement filter (LPC) and said long-term analysis means (LTA, LTR1).

[Claim 30] The means (STR2) for reconfigurating a linearity preliminary announcement multiplier in the short-term analysis means (STA, STR1) and decoder in an encoder The means for performing linear interpolation between the values relevant to two continuing shelf-lives is included about said display in the frequency domain. And equipment of claims 28 or 29 characterized by supplying the interpolated value in the initial PERT of a shelf-life with a multiplier of one set to the short-term composition filter (STS1, STS', STS2) of said **** system.

[Claim 31] The means (LTR2) for reconfigurating a solution parameter over a long period of time in the long-term analysis means (LTA, LTR1) and decoder in an encoder Compare the parameter relevant to two continuing shelf-lives, and the comparison means for generating the flag (F) which makes it possible to perform interpolation between these parameters when they fulfill the conditions decided beforehand is included. The long-term composition filter (LTS1, LTS2) of said 1st and 2nd **** systems When there is said flag, secondary polynomial interpolation of said parameter extended to all the shelf-lives is performed. And the encoder of any one publication of claim 28-30 characterized by a means to supply the parameter interpolated at each long-term composition filter (LTS1, LTS2) being interlocked with.

[Claim 32] The circuit where said time amount shift means (TS) carries out the rise sampling of the residual signal (US), The residual signal sample by which the rise sampling of the 1st group corresponding to the sample of said 1st number LS was carried out to each block of the sample which should be encoded, It is placed before and after said 1st group, respectively, and the residual signal sample with which the rise sampling of two another groups containing many samples

coordinated with the shift by which max was permitted was carried out is memorized. The command by the energy minimization means (EM) is faced. And to the 2nd **** system (STS', STW') The 1st group's thing and the sample of the same number are included. By and said optimal shift The encoder of any one publication of claim 28-31 characterized by including the storage means (SH) for supplying the residual signal sample which was shifted about the 1st group, and by which the rise sampling of the 4th group was carried out.

DETAILED DESCRIPTION

[Detailed Description of the Invention]

[0001]

[Industrial Application] Preferably, the above-mentioned encoder can use this invention by the good engine performance for being concerned with the voice coder which adopted the analysis-composition technique, and specifying further, for example, it is concerned with the encoder which fits the low bit rate application in the minimum of the speed range in 4 - 8 K bit [/] within the limits s. There is a voice coder of the schedule used to the so-called high-speed channel of the Europe mobile radio method as an example of application of this form. In the encoder which uses the analysis-composition technique over each block of the sound signal sample encoded, the excitation signal over the synthetic filter which simulates voice generation equipment is chosen within 1 set of excitation signals so that significant distorted magnitude which may be perceived may be made into min. Generally this is obtained through the comparison of the sample which the original signal in a suitable filter with the function in consideration of the ability of human being's consciousness to evaluate [how] consequent distortion compounded, and a corresponding sample, and instantaneous weighting. In the most general gestalt, the synthetic filter here contains two concatenated components which impose spectrum nature on an excitation signal a short period and over a long period of time, respectively. The former is coordinated with correlation between the continuing samples, and the spectrum envelope which is not flat is generated, and the latter is coordinated with the correlation during the contiguity pitch period for which the fine signal spectrum configuration depends on it. In the case of this method, the signal encoded includes the information relevant to excitation, a short-term composition parameter (a short-term linearity preliminary announcement multiplier or other amounts relevant to them), and a long-term composition parameter (long-term delay and linearity preliminary announcement multiplier).

[0002] Although especially insertion of the long-period-of-time nature to the signal encoded if the delay was updated in each reverse frame in the inside of an analysis-composition cycle greatly improves natural sound of a signal, the related information needs most bits required for coding. It is important to discover the solution which enables reduction of the amount of the information on the schedule transmitted to a decoder, preserving the quality of a signal especially in low bit rate application. 23-March 26, 1992, San Francisco (U.S., California), In the paper I-337 of the name "the application to analysis-composition coding and the pitch preliminary announcement which

were generalized" by W.B. Crane, R.P. Lama Chan Derain, and P. clone which were shown at the ICASSP92 meeting It has suggested performing interpolation by the long-term analysis delay updated with each frame for this purpose. The direct interpolation in those without a suitable array makes a time irregular train cause between the long-term spectral characteristics in an original signal and the compounded original signal, and generates a remarkable distortion while it gives the delay value which is not an optimum value. In order to avoid such un-arranging, in the above-mentioned paper, the long-term preliminary announcement machine parameter became the known function of time amount, and it has suggested correcting the original signal so that direct interpolation to which the engine performance is not reduced may be enabled. The suggested correction consists of time amount vibration and the small amplitude scaling which were limited about the original signal. Time amount vibration is performed in a discrete mode. The need for setting up the optimal amount, in order to insert such a time amount vibration therefore increases the complexity of an encoder clearly.

[0003] Therefore, in order to solve this problem, in this invention, before long-term analysis, the time amount shift according to individual is introduced on that residual signal, and the coding system performed so that the search to the optimal excitation signal and the optimal shift may mitigate the complexity of count is used. The description of this invention is indicated in the attached claim. The desirable example of this invention will be described with reference to an attachment **** drawing. In an accompanying drawing, : drawing 1 is the block diagram of an encoder,; drawing 2 is the functional diagram of a certain block of an encoder, and; and drawing 3 are the block diagrams of a decoder. Before describing an encoder / decoder structure in a detail, it summarizes a basic principle. several [which the encoder was encoded and was fixed] -- the sample x (n) of the sound signal of a schedule packed into the block (generally called a "frame") containing the sample which Lf follows is received. Each frame of Lf sample is divided after that to the reverse frame of the continuation **** sample of Ls. An encoder will determine the one-set parameter which should be transmitted to the decoder so that the signal with which a decoder approximates the original signal can be compounded. In order to attain this, an analysis-synthesis procedure is used, it lets it pass, an encoder analyzes the effect of the possible value of each parameter, and the value which makes it possible to obtain the best approximation about the original signal is chosen. For this reason, the encoder contains the replica of that decoder, in order to make a corresponding output signal to each of said value. Since such an output signal is generated, both the ** term of a sound signal and short-term correlation are used, and it is imposed on an excitation signal through each synthetic filter. In each frame, an encoder performs linearity preliminary announcement analysis (a short period or LPC analysis), and calculates the short-term residual signal used in order to calculate the parameter (delay and multiplier) of a long-term composition filter. (Since the primary filter is used, that multiplier is peculiar in this desirable example) . In order to improve the resolution of long-term correlation information, delay of a frame present in the delay and multiplier both and a front frame is interpolated in a value, when near. In order to decrease the effect of the time mismatching between the signal of the origin of it, and the reconfigurated signal, in each

reverse frame, a small time shift is introduced into the original sound signal, and it is determined through a thorough search in the range of a possible value that the shift amount will make min the error, i.e., the energy of an error (difference between the signal of a dimension and the signal reconfigurated). The search to the optimal excitation signal is performed after determining the optimal shift.

[0004] In order to make a publication still clearer, in the following explanation the possible excitation signal It is treated as a word chosen with the definite **** sign document for the encoder of the form known as CELP (sign document excitation linearity preliminary announcement). This It is **** even noting that it consists of a very small number accompanied by the amplitude and location where each word was defined beforehand deterministically temporarily of pulses (preferably 1 or 2) and does not have a sign document. The encoded signal includes in usual the short period and long-term composition filter parameter which are transmitted in the gestalt of the index encoded suitably, and the information relevant to the optimal excitation. In a decoder, such an index is left, in order to give the signal which the excitation signal corresponding to what was used by the encoder was searched, and was reconfigurated, it **** in a series of long-term composition filters and short-term composition filters, and the reconfigurated signal will receive much more ******** (ex post facto ****) based on a short-term composition parameter, in order to improve a subjective signal quality still more for example. Then, the reconfigurated signal is again changed into an analog gestalt, and is supplied to use equipment. As an example, by the following publications, the examination to the frame of the sample (in the sampling frequency of 8kHz, it corresponds to the sound signal segment the die length of whose is T= 20ms) of die-length Lf=160 is made, and, as for the frame, die length Ls is divided to eight reverse frames of 20 samples. It is required to use the HfK sample (for example, H= 24, K= 8) as a group of the following frames for the reason relevant to installation of a time amount shift in addition to Lf sample of a frame.

[0005] For referring to drawing 1 , it memorizes temporarily, and for every Tms, it writes in and reads to the buffer MT arranged so that the input signal sample x on Rhine 1 (n) might memorize the sample of N=Lf+H+K, and Lf sample as a block is taken out. The sample read in MT is supplied to high-pass-filter FPA which excepts a direct-current drift and a low frequency noise, and the ****(ed) signal Xf (n) is supplied to the short-term analysis circuit STA and the linearity preliminary announcement filter LPC. To each frame, Circuit STA determines P linearity preliminary announcement multiplier ai (for example, 10) of one set, changes such a multiplier into the parameter of one group in the frequency domain generally known as LSP (Rhine spectrum pair), and performs the quantization 1 of the difference between adjoining parameters, for example, a scaler. The index j (phi) which is a part of encoded signal is transmitted to a decoder through Rhine 2a, after binary encoding in the circuit which is not illustrated. Since the spectrum line has the property of quantization of synthetic filter stability better than the thing of a multiplier, interpolation, and a check as everyone knows, the conversion to the Rhine spectrum pair is desirable. smoothing of the spectrum information relevant to a characteristic frequency region by the block STA before calculating the Rhine spectrum pair -- it -- a quantization circuit -- also in

order to adjust resolution, it performs. This is the calculated multiplier ai. Each factor gamma1 i It is attained by multiplying, and as a criterion, although the value is smaller than 1, it is close to about 1. Especially, in the case of a narrow characteristic frequency region, although this actuation is equally narrow, the danger of reproducing after quantizing the characteristic frequency region shifted about the original thing is decreased, and, so, the cause over degradation of the quality of the encoded signal is decreased.

[0006] Circuit STA follows "digital signal processing about a sound signal" by L.R. RABINA and R.W. Shaffer (cousin-hole Ed., en gel wood KURIFFU, N.J., USA, 1978), and the conventional automatic correlation technique which is described by 401 pages, and is a multiplier ai. It calculates. STA is an one-set Lf+P input sample (especially) obtained through the trapezoidal shape aperture which carries out weighting to it being also at the greatest weight (especially 1) about all the samples except P sample of the beginning and the last to count. It operates on the sample which occupies the Lf+P location of the last in MT. It is determined that the weight to it is also for the easy linear interpolation actuation between the minimum weight and the maximum weight, and smoothing needed for giving a good result to this appearance with an automatic correlation technique is restricted to the duplication field between the continuing apertures. Moreover, when positioning in the forward direction of the aperture encodes the first reverse frame (three [for example,] of the beginnings) of a frame instead of the preliminary announcement multiplier of the linearity calculated to the frame itself, The fact that the multiplier obtained by conversion of the Rhine spectrum pair value determined through interpolation between the value relevant to a front frame and the value relevant to the present frame is used is taken into consideration. This has guaranteed the ****** shift between the parameter of the present frame, and the parameter of a front frame. It is voice and the IEEE minutes about signal processing, and a paper of "count of the Rhine spectrum frequency which used the CHIEBISHIEBU polynomial", and conversion of the linearity preliminary announcement multiplier to the Rhine spectrum pair will be performed in December, 1986 in acoustical and the approach proposed by P. KABARU and R.P. Lama Chan Derain, for example. Probably, detailed description will be unnecessary since actuation of STA is a standard linearity preliminary announcement encoder.

[0007] Moreover, Index j (phi) is supplied to the linearity preliminary announcement multiplier reconstruction circuit STR 1, and supplies the value quantized about the multiplier obtained by applying a reverse procedure about what was used for a circuit STR 1 changing the multiplier into the Rhine spectrum pair at Filter LPC, the short-term composition filter STS 1, STS' and the spectrum weighting filter SW, and SW'. Furthermore, STR1 also calculates the interpolated value which should be used by the first three reverse frames. The value quantized in the following publications is ai because of simplification. It is specified. Filter LPC receives ****(ed) sound signal sample xf (n), and is ordinary function 1-A (z) (however, [Equation 1]) about them.

It is alike, it follows and **** and is the short-term preliminary announcement residual signal rs. (n) is generated and it is the residual signal rs. (n) is the ****(ed) residual signal rf. The low-pass filter FPB which makes (n), and corrected residual signal rm The time amount shift circuit TS which makes (n) is supplied. Low-pass ******** makes easy actuation of the following long-term analysis circuits LTA as everyone knows. Circuit LTA is determined in each frame and supplies, Gain b, i.e., the multiplier, by which weighting of the sample is carried out to the delay d (pitch period) used since the signal with which the sample of an excitation signal was reconfigurated by the following long-term composition filter LTS1 is generated in the first half.
[0008] as the presetting number which took the die length of the aperture into consideration for the count which makes it possible to change Block LTA between the minimum values and maximums which were allowed k by Delay d (for example, 20 and 120), and to acquire a satisfactory value [as opposed to d for x] -- the following automatic correlation function, i.e., :, -- [Equation 2]

Delay d is calculated by making it max. As it already said that it takes into consideration that a sample with the newest aperture must be included, the die length is compromise between two opposite requirements, and its precision of the evaluation is also so large that die length is large. On the other hand, the aperture is enabled to acquire the present value made to adjoin the edge which interpolation takes it, since a short ******** shorter paddle approaches the edge of a frame at which the core should be encoded (Lf sample). For example, x is set to K. In a desirable example, the delay never becomes below the die length of a reverse frame, but this greatly simplifies the continuing actuation. Moreover, it is corrected, the value calculated by the formula (1) is investigated after that, it guarantees the smooth configuration over d as it can do, and it amends the synchronous loss by time amount shift. the value of a multiplier b -- a degree type, i.e., :, -- [Equation 3]
r1 (n) =rf (n)-b-rf (n-d) (2)
Error signal r1 in the output of LTS1 which is alike and is given more Making energy of (n) into min is determined. For b, E (rf) is a degree type to the delay value d used to the present frame.;
[Equation 4]

$$E\ (r_f) = \sum_{n=x}^{N-1-d} r_f^{\ 2}\ (n) \qquad (3)$$

It is given by formula b=R[rf (d)] / E (rf) noting that it comes out and the energy which becomes settled is shown. To the value of b, the minimum value 0 and maximum 1 are set up, respectively. Although it is excepted since the reverse signal it forces that zero or less value transmits a notation bit is supported, one or more values make a filter unstable as everyone knows. The value of b calculated using the formula (2) is corrected so that the best quality of the encoded signal may be guaranteed. Furthermore, it is possible to use the value acquired by the linear interpolation between the value calculated to the front frame with a certain frame instead of the values d and b

calculated by the formula (1) and (2) and the value calculated to the present frame.
[0009] With count of d and b, the preliminary announcement gain G was calculated, and it is an amount showing the ratio between the input from a long-term preliminary announcement machine, and the energy of an output signal, and has given the magnitude of long-term preliminary announcement effectiveness. Gain G is a formula [several 5].

$$G = \frac{1}{1 - bR[r_f(d)]/E'(r_f)}$$

it is specified be alike -- having -- an upper type -- :[Equation 6]

$$E'(r_f) = \sum_{n=X+d}^{N-1} r_f^{\,2}(n) \qquad (4)$$

For Gain G, it is used for deciding whether to be that the voice segment currently encoded is pronounced, and it is each threshold Gthr. It is shown by the larger value of G and b than bthr. In the case of the sound uttered, LTA performs interpolation and generates the flag V used in order to determine to introduce a time amount shift. If the first correction to Delay d is due to the search to the maximum of a function [/ near / predetermined / the value acquired with a front frame (for example, **15%)] (1) and the maximums of a there differ from maximums only with a main amount smaller than a certain limit, the new value which gives a convenient and much more smooth appearance to interpolation will be used. This secondary search is performed, only when the signal in a front frame is pronounced strongly and interpolation is received. Furthermore, the correction in the case of being carried out is made, before calculating b and G so that the value already corrected about d may be used to such count.
[0010] The 2nd correction is adjustable delay [outside 1] by which the effect is compared with the thing of asynchronous operation of an encoder.

$\hat{h}$

It is coordinated with existence of the time amount shift device to insert. In order to recover synchronicity, it is calculated by LTA and the value of d corrected as said before is coordinated with the amount of the shift [itself], and it is [Equation 7].

$$d' = \hat{h}\, d / \Gamma L f$$

It is changed by adding correction term d' given as be alike to it, it sets at a front ceremony, and is [External Character 2].

$\hat{h}$

It is the shift accumulated to the frame with which it is expressed as the number of the samples of the residual signal raised and sampled by ****** gamma, and d and Lf have the same semantics as having stated above. A rise sampling will be further discussed by the detail with reference to Circuit

TS. Correction needs interpolation for the present frame, and when a voice segment is not uttered, it is performed. the first conditions do not have interpolation -- if it becomes, since a shift will not be performed -- required -- moreover, this situation -- setting -- an exact value -- Seki -- since the minimum uniform correction of d is usually perceived the bottom, the above-mentioned signal is not uttered. The absolute value is restricted to maximum |d'| max called 1 before adding a correction term to d. Furthermore, the correction does not correct the decision (described also later) about interpolation, but only when not taking besides the value of the range which was able to give the value of d, it is performed. If the first correction has too large b about b, since increase of too much energy will arise and a noise will be produced, it is the 1st upper limit b1. It consists of clipping b. limit b1 it coordinates with the ratio between the energy in the pitch period of the frame of the present frame and a front -- having -- it -- a degree type --;b1 =[E "(rf) 0/E" (rf)-1] d/2Lf it gives -- having -- an upper type -- setting -- E " (rf) -- amount [several 8]

$$\sum_{N-1-d}^{N-1} r_f^{\,2}(n)$$

******(ing), it is the energy in the pitch period d, and an index 0 and -1 express the present frame and the front frame, respectively. Correction is made when the energy in a front frame exceeds a certain threshold.

[0011] If the voice segment in which the value of G has low (below Gthr) and low periodicity is shown, b will be performed when comparatively large (larger than the 2nd limit b2), and an actual value is adopted in this case, since an artifact will be made to the encoded signal, much more limit to b is a value b2. It is adopted. When talking about interpolation, as for this, two relative fluctuation of continuing inter-frame d does not exceed the amount (for example, 15%) decided beforehand as an absolute value, but when both the values of b in such a frame are forward, it performs. Actual count of the value of d and b which are used in interpolation is performed in the long-term composition filter LTS1 with which LTA sends Flag F, when the above-mentioned conditions are collated. Moreover, the same flag is supplied to the circuit EM which determines the optimal time amount shift and excitation. Although the information about interpolation is needed by the synthetic filter in a decoder, since it can remake immediately in the filter, it is not necessary like [in an encoder] to transmit it exactly by the comparison between the values of d and b relevant to two frames. The value of d and b which are determined with each frame is the information relevant to the long-term analysis of the schedule inserted in the encoded signal like [in before], and is changed into each index j (d) transmitted to a decoder through Rhine 2b and 2c after suitable coding, and j (b). Index j (b) is determined through quantization actuation, in the meantime, maximum is restricted to 1 and also the value of b below one half of a value where the beginning was quantized is set to 0. However, since d is already discrete ******, the quantization about d is not required and it is desirable to transmit d with the gestalt of an index for [with other information] homogeneity. conversion on the index of the value of d -- actually -- the range of a possible value -- value dmin from -- instead, it consists of those shifts that begin from 1. In the

described example (101 values of d and j (d)), 7 bits is required to encode index j (d), and such a bit enables coding of the value of j (d) in the outside of the predetermined range. It is used for it being shown that one of such the much more values (for example, value 127) sets b to 0, and does not contribute to the signal with which the long-term composition filter was reconfigurated if it was b= 0, but since delay information is unnecessary, a decoder is supplied instead of index j (d) corresponding to the actual value of d. However, index j (b) corresponding to the minimum value of b is transmitted besides the information which forces 0 into b.

[0012] The circuit which generates index j (b) and j (d) for simplification is included in Block LTA. Since Circuit LTA can make a decision relevant to the property and need for a sound of performing interpolation, i.e., a shift, only depending on the value with which b was corrected, notice the correction about d in consideration of a possible shift about the point performed after correction of b. The actuation performed by LTA is described by the detail with the appendix including the program written by C. Since it is written, an engineer is satisfactory at the point of designing the equipment which performs the described function. It is got blocked and the reconstruction circuit LTR 1 containing the easy read only memory addressed by the index reconverts index j (d) and j (b) to the reconfigurated value for which each parameter was quantized. This reconstruction working and LTR1 give the actual value of d and b, when the value (that is, j (d) is in the range of 1-101) as which j (d) considered delay is shown. They are the value [as opposed to / if any one of the values (the value is 102 to 127) out of range allowed j (d) is shown / b in LTR1] 0, and the value dmin over d. It gives. When reconfigurating a parameter, as for the fact that all indexes j (d) which is not what does not correspond to the value in consideration of that delay, and is actually used for this purpose forces 0 into b and of being interpreted as a display, even the case of the error by the least significant bit of that index makes it possible to reconfigurate a value b= 0. Anyhow, by chance, if reconstruction of b= 0 goes wrong, since the circuit LTR 1 has index j (b) to which they correspond on the occasion of the use, the minimum value of b is generated. The value reconfigurated below for simplification (that is, quantization) is shown as b and d.

[0013] The long-term composition filter LTS1 follows ordinary 1/[of functions] P(z) =1/1-b-z-d, and is the excitation signal s1. Short-term residual signal ss reconfigurated by ****(ing) (n) (n) is generated. This consists of the gestalt information (renovation) expressed using the amplitude parameter g (renovation gain) of forward [which was chosen with the sign document of the renovation gain IG 1 by one of a word andthe s (n) of the renovation sign document IC 1], or zero, and the notation information the value is indicated to be with the parameter sigma (renovation notation) which is **1. Therefore, signal s1 (n) is s1. It is given by (n) = sigma-g-s (n) =g1 and s (n), and is obtained through a multiplier M1. Parameter sigma shall be read in the sign document IG 1 for simplification. In order to make an understanding easy, as stated above that the sign documents IC1 and IG1 were represented as a circuit block (the memory containing them is suggested), the specific structure of a renovation sign document makes those storage an extraneous article temporarily. The structure of renovation and a gain sign document is examined later. Reconfigurated residual signal ss Weighting must be carried out to it being also at Factor b about

the sample relevant to [in order to obtain the sample of (n) / at the moment] n-d in LTS1. When interpolation is not performed, actuation of LTS1 is completely the same as the conventional thing. the case of interpolation -- the value of d and b -- n=0 -- as Lf-1, deltad= [d0-d (-1)]/Lf, and deltab= [b0-b (-1)]/Lf -- a degree type, i.e., :, -- [Equation 9]

$$d(n) = d(-1) + (n+1)\Delta d$$
$$b(n) = b(-1) + (n+1)\Delta b \qquad (5)$$

It is alike, and it is calculated for every sample by following. Notations d0 and b0 show the value relevant to the present frame, and d (-1) and b (-1) show the value relevant to a front frame. Therefore, interpolation becomes linearity-like and extends over all frames. The value of d (n) and b (n) changes for every sample. It is the signal [in / d / (n) / generally / in not an integer but this / moment n-d (n) in continuation **** time amount] ss. The value of (n) is not in agreement with the thing of the sample which can actually be used, and means what must be evaluated. according to this invention, this evaluation is performed through the secondary polynomial interpolation which has a core at the moment of the discrete **** time amount nearest to n-d (n) (namely, a parabola -- letting it pass), and interpolated value b (n) is multiplied by the value evaluated as it thinks best. Count with the interpolation procedure adopted is farther [than the advanced interpolation based on ****(ing) a signal] easy. However, the effectiveness is substantially the same as a low-pass filter, and since it avoids having the periodicity which the reconfigurated signal ******** too much, it is useful for good actuation of an encoder.

[0014] Reconfigurated short-term residual signal ss (n) is supplied to the short-term composition filter STS 1 with the transfer function of 1 / 1-A (z). This filter generates reconfigurated sound signal y (n) which is supplied to the spectrum weighting filter SW with the transfer function of [1-A(z)]/[1-Aw(z)] as usual. Setting at a front ceremony, Aw (z) is a function [several 10].

$$\sum_{i=1}^{p} a_{wi} \cdot z^{-i}$$

Coming out, awi=ai gammai and gamma are correction factors which determine the band which extends near a characteristic frequency region and which are determined experimentally. Signal yw by which weighting was reconfigurated and carried out (n) is the correcting signal xw which was acquired by ****(ing) the output signal from TS in respectively same filter STS[of two concatenation ****] ', and SW' to STS1 and SW and by which weighting was reconfigurated and carried out. In Adder SM, it is deducted from (n). With the output of SM, error signal e (n) by which weighting was carried out is obtained, and the signal is supplied to the error energy minimization circuit EM which performs all required actuation, in order to determine the optimal shift and excitation.

[0015] The purpose of Circuit TS is aligning the signal of the schedule encoded as the replica which a long-term composition filter's makes is also in time amount, and is avoiding the shift between the

pitch peaks in the signal especially announced beforehand by LTS1 and the original signal. For this reason, only a certain decision ******deltah will shift the time window of Ls sample at which TS in each reverse frame positions that reverse frame itself. It is determined by Unit EM that the shift of the schedule applied is also by the high-speed search procedure in within the limits of the value specified by the shift which can permit max. Although a shift with a residual signal is applied since a consequent distortion becomes what continuing ******** in STS' and SW' smooths and cannot be substantially sensed by it, a shift with the original signal is not applied. The shift applied by the reverse frame is algebraically added to what was accumulated by the time amount, and in order to avoid too rapid fluctuation, it gives a shift on the whole. A shift on the whole cannot exceed a certain maximum (H sample of the signal of a dimension). Therefore, the reason of to why H sample of the frame which follows is loaded by MT is clear. The purpose which restricts shift fluctuation is avoiding too much distortion, and, so, the delay which must be permitted in a coding procedure opts for the limit relevant to a shift on the whole by the availability of a future sample. The time amount shift has resolution smaller than one sampling period of the original signal, and it is required to perform the rise sampling of a residual signal so. In consideration of these all, Circuit TS is the residual signal by which the rise sampling was carried out at the output.

[External Character 3]

$\hat{r}_s(\hat{n})$

****** rise sampling circuit US (in fact interpolation filter) and a shift entity [outside 4]

$\hat{h}$

The residual signal which received from EM and corrected the information boiled and attached and by which the rise sampling was carried out

[External Character 5]

$\hat{r}_m(\hat{n})$

The component SH for a shift to generate is included. As an example, rise sampling-ized gamma is 8, so, the signal by which the rise sampling was carried out has the frequency of 64kHz, and this rise sampling ratio gives suitable resolution to the purpose of all requests. Furthermore, for right actuation of an interpolation filter, it is required to always use the sample of a predetermined number following that interested, and this is the reason K more samples of the continuing frame are loaded also to MT.

[0016] It is not necessary to carry out substantially that it is also at the sampling frequency of 8-KHz about the down sampling which acquires the corrected residual signal. saying -- the actuation -- the need -- responding -- it is also at a suitable phase -- a ratio -- every gamma --

[External Character 6]

$\hat{r}_m(\hat{n})$

By only reading a ** sample, it is because it performs tacitly. gammaLs sample of the residual signal with which the rise sampling of the component SH was carried out as a practical question, The sample of a certain settled number of degrees coordinated with the shift by which the max in a frame was permitted, and a front (in fact) As explained by the publication about the optimal shift search, it is the memory which loads a number of the maximum shift equal twice of samples to each reverse frame. And the component SH It is addressed for read-out by the error energy minimization unit EM so that Ls sample appropriately shifted to future circuits about the coming reverse frame may be supplied. Each has Ls sample to examine a renovation sign document, and this includes the word of the predetermined number in which some of them differ from 0 in it. Thinking that this selection can find out the word which has many pulses (namely, sample which is not zero) for which all pulses are actually suitable there since a sign document has constraint makes it possible to decrease the amount of required count, when it originates in the fact of being a phantasm and the optimal excitation is discovered. In the desirable example of this invention, the sign document consists of the two sections. The first section includes Ls language with the amplitude equal to 1, the sample which is not the single zero which have a plus sign, and the sample of the zero of Ls-1. The sample which is not zero occupies a different location in all the words acquired one [at a time], when only one location only shifts the sample which is not zero. Signal S (n) is [Equation 11] to this first section of a sign document.

s (n) =delta (n-n1) (5)

It is expressed by carrying out, delta is a well-known unitary function by the upper formula, and it is n and n1. It can have a value between 0 and Ls-1.

[0017] The 2nd section includes the word which has the sample which is two whose amplitude is 1, and the zero sample of Ls-2. Such a word leaves a number of keywords limited as it is also with the approach indicated by European Patent application EP-A -0396121 in a name called CSELT, and is generated. In the example currently taken into consideration all of three keywords The 1st pulse in a location 0, Each key position n2 (1), n2 (2), and n2 It has the 2nd pulse of (3)., and other WORD, i.e., word It is obtained in shifting a pulse pair towards the end of the word until the 2nd pulse reaches in the end of a word or the 1st pulse arrives at each key position. Only the 2nd pulse notation has two mutually different words, and they take the total of the word in a renovation sign document to Ls+2nickel2 (this example 62) as a key position is chosen in order to give the origin to the possible location of nickel2 (especially 21) of that pulse pair, and described by the above-mentioned Europe application to one each of such the locations. this 2nd section of a sign document -- receiving -- a renovation word -- n= 0 .... Ls- 1 and n1 =0 ..Ls-1-n2 (p) n2 =n2 (p) .. Ls- 1, p=1 ..Nip, and n2 as the number of the key positions where (p) shows a general key position and Nip is used (this example 3) -- a degree type, i.e., :, -- [Equation 12]

s (n) =delta(n-n1) **delta (n-n2) (6)

It is expressed be alike. The renovation sign document structure with the sample and word which are not some zero obtained when a limited number of keys are left and only one location shifts a sample is the easy deterministic structure where storage of a sign document also makes possible

the quick search procedure of the optimal excitation which does not need effective **** of a candidate excitation signal, either.

[0018] During the search to the optimal renovation, the trial by it being also in the word of the first section of a sign document must be performed, only when the time of long-term analysis displaying an utterance sound or energy concentration strong on the contrary is seen in the short signal section. Since such strong concentration can sign, the initiation, i.e., the standup, of the section uttered, the classification is based on long-term analysis still more and there are no features which are helpful for this standup being shown at the last signal section when actual, as they thinks best, they are classified. Therefore, a filter LTS1 can supply a right preliminary announcement signal under such a condition. then, it comes out absolutely that a pitch pulse is reproduced correctly, and there is, and so, although unsuitable actuation (in utterance section) or the impossible corrective action (starting and coming out) of a long-term composition filter is compensated, it is useful [the thing / use of a single pulse word] for the signal with which good quality was encoded, in itself. It is not used for instead a single pulse word reproducing the sound which does not start and come out and which is not uttered, but since subjective effect is bad, even when it is actually one of them to give the minimum error signal energy, use of a single pulse word is usually an opposite effect. The approach by which the strong energy concentration in short time amount is detected will be described later. The word in a sign document is checked by each index j (s), and the index relevant to the optimal word encoded appropriately is transmitted to a decoder through Rhine 2d. In an example here, since indexes [many] j (s) supports those words including 62 words, a sign document can use two still more nearly another values of j (s) which does not support any words in the sign document, without correcting several j (s) of bit coding. If these are used for expressing the renovation gain of zero and are referred to as what is similarly made to long-term preliminary announcement delay and a multiplier so that it may be stated also later It is used for only one of the two values of j (s) which does not support a renovation word displaying g= 0, when generating such an index, and when decoding, g will **** in both the values of j (s), and it will be set as 0.

[0019] This is quantized about Gain g using the sign document made so that it might permit saving a coding bit about a thing actually required expressing all the possible values given to the sign document. The information about gain over each reverse frame is expressed in the gestalt of two indexes j (gmax) and j (gnor), and the thing of the start of them is coordinated with the maximum of g in a frame, and the 2nd thing is coordinated with the difference between this maximum and an actual value with Notation sigma. This information is transmitted to a decoder through Rhine 2e. The sign document contains the possible absolute value of the Nig individual of g expressed as Nig=Nim+Nin -1 noting that Nin shows two different powers, Nim and2. Here, it is Nim=24, Nin=22 or Nim=24, and Nin=23. It can have. In each reverse frame, the optimum value of g determined that it is also by the error minimization procedure described later is quantized, and although not transmitted, each index j (g) reconfigurated with a decoder is generated. In the end of a frame, the value j relevant to the greatest frame gain (gmax) is identified, and when it is below Nin, it is transmitted. Otherwise, Index j (gmax) is made into a value Nin. Thus, j (gmax) can take a

Nim value and, so, the number which is a coding bit is restricted. Supposing it identifies j (gmax), it is calculated to each reverse frame that Index j (gnor) is also for formula j(gnor) =j(gmax)-j (g), and j (gnor) can once have a value in the range between 0 and Nim+Nin -2. The actual value of Index j (gnor) is transmitted only when not larger than Nin-1. When there is nothing as if, gain is ****(ed) with 0 (that is, renovation becomes silent to a reverse frame and the gain is very small about maximum), and index [of a renovation word] j (s) is made into one of the values which do not correspond to which [which show transfer of the word of zero gain] sign ****. Thus, the bit which the converted differential dynamics were used and was used for expressing the gain in all dynamics is saved at the sacrifice of the slight engine-performance loss by the renovation silence which may take place. In order to make effect of the channel error about renovation index j (s) into min, in silence, value Nin-1 to Index j (gnor) will be transmitted anyhow.

[0020] a gain sign document -- a logarithm -- since it is a sign document, the ratio between two continuing values is fixed. This ratio is taking into consideration some requirements shown below. That is, the value in :-dB must be near as much as possible, in order to make quantization as exact as possible.

- The dynamics on the whole between minimum gain g (1) and the maximum gain g (Nim+Nin -1) must be appropriately extended so that the voice level from which the sound of a different form and a suitable set differ may be covered.

- The differential dynamics between g (x-Nim +1) (x) must be appropriately extended, in order to make possibility of silence low suitably. For example, in the case of the value which Nim and Nin mentioned above, the value of the ratio between two continuing gain level is continued for 6dB from 3dB. Now, here describes the high-speed search procedure to the optimal shift and excitation with reference to the operation Fig. in drawing 2 corresponding to the set of the blocks M1, LTS, and STS of Fig. 1, STS', SM and SW, and SW'. The blocks STW1 and STW2 by which the filter produced from a series of filters STS1 and SW is expressed with drawing 2 , and transfer function 1/1-Aw Except for block STS' which is a filter with (z), and SW', the same notation as the thing in drawing 1 is used. the component (LTSa, STW1a, STW2a) which has the zero input whose each of a filter gives the contribution of initial condition (memory to a front reverse frame is ****(ed)) in this drawing -- and it is divided to the component (STW1b, STW2b) reset in each reverse frame (it is **** at the initial condition of zero) as shown by the signal R supplied by the hourly base which is not shown. Since Delay d is assumed to be smaller than a reverse frame, **** of being also at the initial condition of the zero about excitation is only short-term ********.

[0021] The decision of the optimal shift is evaluation of the need of performing the following three steps, i.e., :-shift.;

- Decision of the suitable range about a shift value;

- It consists of searches to the optimal shift in the range. the first step -- three conditions, i.e.,;

- A reverse frame is rs. It is not the silence the energy of (n) is indicated to be according to the fact of being larger than a predetermined threshold.;

- A signal is uttered as shown by the flags F and V from LTA, or it is interpolated.;

- rs It is actually generated in a reverse frame and the peak of (n) is rs in a reverse frame. The mean power (several [of a sample] energy broken by Ls) of (n) is larger than the energy in the period of die-length d finished as the sample of the last of the reverse frame itself, or is shown by the fact of being equal to it. It is confirmed whether to be that the conditions to say are fulfilled. The reason over the first conditions is clear. About the 2nd and 3rd conditions, a shift must be performed, only when a pitch peak is in a reverse frame. The fact that produced this in the utterance section in the 1st first, and interpolation arose, That is, since the fact that the value of the parameter obtained with two continuing frames is very near has suggested the positive periodicity in the signal segment which must be encoded In this case, it is useful to enable that shift, although the irregular danger that it can set between the reconfigurated signal and the original signal is made still smaller. Count of energy and power is separately performed about a rise sampling signal or the original signal. [External Character 7] in the reverse frame present in such count

$\hat{r}_s$

A ****** absolute value and its location are also obtained, and it is used in case the shift is determined. In order to decide the location about maximum and to obtain the greatest resolution, it is absolute to make it operate about the signal by which the rise sampling was carried out.

[0022]

[External Character 8]

第２のステップは、フレームにそれまでに累積されたシフト値$\hat{h}$の周囲に延在する範囲の下限および上限$\hat{h}_{min}$，$\hat{h}_{max}$を決定する。値$\hat{h}_{max}$，$\hat{h}_{min}$は、差$\hat{h}_{max}$－$\hat{h}$および$\hat{h}$－$\hat{h}_{min}$が、例えば、アップサンプリングされた信号$\hat{r}_s$の２０サンプルという前以って規定された値Γ・Δｈを持つように、初めに定められる。それ故、最適のシフトがその中から捜索される最大数の可能な値（この例では４１）が存在する。実際の極値$\hat{h}_{min}$，$\hat{h}_{max}$は、過去においては前以って考慮された$\hat{r}_s$の最大の可能な複製で、そして将来においては最大値の結果的損失でもって、副フレームをシフトさせ過ぎるのを回避させる必要があるので、値$\hat{h}$に関して対称でない（すなわち、その範囲は累積された値$\hat{h}$の片側または両側に制限される）。このチェックは、$\hat{r}_s$の最大値を副フレームに記憶することによって可能になる。しかしながら、範囲の制限が両側でない限り、最適なシフトに対する捜索は、制約を受けない極値を超えた或る値を考慮することにより、その範囲の幅を一定に保つように実行される。いかなる場合にも、実行される予定のシフトは値Ｈを超えてはならない。

The optimal shift value in trial within the limits makes min energy of an error signal e1 (n)

expressed with the difference between the contributions yw1 (n) of the excitation which **** the correcting signal xw (n) by which weighting was reconfigurated and carried out, (drawing 1), and memory, and is acquired as it is also by the high-speed search procedure of decreasing required computational complexity. On the other hand, it is the output signal xw from STW' to this high-speed search. (n) is [Equation 13].

$$x_w(n) = \hat{r}_m(\hat{n}) + \sum_{i=1}^{p} a_{wi} \cdot x_w(n-$$

$$= \hat{r}_s(\Gamma n + \hat{h}) + \sum_{i=1}^{p} a_{wi} \cdot x_w$$

(however, n -- from 0 up to Ls-1)

It must carry out and must take into consideration being expressed and that the same signal is the sum of the output xw1 of STW2a, and the output xw2 of STW2b on another side. The sum in a formula (7) expresses the signal xw1 calculated only at once like the contribution yw1 to which Chain LTSa and STW1a correspond, so, the error specified as e0 =xw1-yw1 is also calculated only at once, and the result of the count appears in the output of Adder SMa.

[External Character 9]

その後、エラーe_1は$e_1 = e_0 + x_{w2}$として書かれ、上式でx_{w2}は$\hat{r}_s$、つまり、そのシフトに依存している。次いで、シフトのすべての値に対してx_{w2}を決定し、e_1のそれぞれのエネルギを各々に対して計算し、最小のエネルギと対応する信号$x_w(n)$とを与える$\hat{h}$の値を記憶することが必要である。

[0023] the shift value to which the procedure of determining xw2 adopted according to this invention was given -- receiving -- a signal xw2 -- a degree type, i.e., :, -- [Equation 14]

$$x_{w2}(n) = \hat{r}_s(\Gamma n + \hat{h}) + \sum_{i=1}^{\min(n,p)} a_{wi} \cdot x_w(n-i) \qquad (8)$$

It is taking that be alike is given into consideration. Since a sample with n-k<0, i.e., the sample of a front reverse frame, must not be taken into consideration on the occasion of **** that a peace upper limit is also at the initial condition of zero, it is the minimum value between n and p.

[External Character 10]

実際の場合、x_{w2}の値は、$\hat{h}_{max}$から$\hat{h}_{max}-\Gamma$
Γの可能なシフトに対して式（８）に従って計算
シフトを調べる前に、偶然にして、$\hat{h}_{min}$が到達
になる。$\hat{h}_{max}-\Gamma$から$\hat{h}_{min}$までのシフトの他

instead of calculating that a formula (8) is also -- xw2 -- a degree type, i.e.,;
[Equation 15]

$$x_{w2}(n)=x_{w2}(n-1)+Q(n)\hat{r}_s$$

（但し、$n=Ls-1\cdots\cdots1$, $x_{w2}(0)=\hat{r}_s$

It is alike, and is followed and calculated. In the formula (9), Q (n) shows the pulse response of ****** of Filter STW as Q(0) =1 (since it is calculated only to Ls value of n). When Q takes into consideration what it opts for only at once except for a certain value so that it may understand from here, a formula (9) needs count far fewer than a formula (8).

[0024] When actual, gamma value of xw2 must be calculated according to a formula (8) and (9) to each of the signal sample by which the rise sampling of gamma corresponding to a 8kHz sampling period was carried out. Once it makes energy of e1 (n) into min and finds out the optimal shift, it can begin, in order that minimization of the energy of e (n) may find out the optimal excitation. Unit EM calculates direct the formula of the energy which is the function of the location of the pulse in a renovation word and which should be minimized, and pulse response Q is adopted for this purpose, and it is calculated during the search to the optimal shift. Count of a pulse response is conveniently made about activation of ******** according to the fact that the sample there are many each words and they are not [sample] two zero is included. Furthermore, consideration of the case of a word with two pulses of being more general obtains simply the response to the word of everything but all that are the sum of two responses and were coordinated with the key by which only distance with the pulse response [on the whole] equal to a key was ****(ed) by changing only one sample at once. For simplification, the adjustable range of the sum index to the total extended to all the samples in a reverse frame is not displayed by the following mathematical expressions. error e (n) to a general excitation word -- u (n) -- as the output signal from STW1b -- e(n) =e1(n)- it is given by yw2(n)

=e1(n)-g1 and u (n). the energy of e (n) -- a degree type, i.e., :, -- [Equation 16]
E(e) =sigmae2(n) =sigma [e1(n)-g1 and u (n)] 2 (10)
Be alike is given and it is E(e) =sigmae1 2+2g1, sigmae1 and u+g1 2, and sigmau2. It can write by carrying out. If Display R (e1 u) is used in consideration of the sum of the beginning and the last expressing a signal e1 and the energy of u, and expressing cross-correlation [between them by which the 2nd thing was evaluated to K= 0] R (e1 u) (K);
[Equation 17]
E(e)=E(e1)-2g1 R(e1 u)+g1 2 E(u) (11)
******.
[0025] making E (n) into min -- the difference of energy, i.e., :, -- [Equation 18]
deltaE=E(e1)-E(e) =2g1 R(e1 u)-g1 2 E (u) (12)
It is the same as making it max. each word of the consulted sign document -- receiving -- the max of a formula (12) -- g1 g1 which appears immediately by calculating the derivative by being related and setting it to 0 value g0 = -- it obtains to R (e1 u)/E (u) -- having -- it -- receiving -- a degree type, i.e., :, -- [Equation 19]
deltaE0 =R(e1 u)2/E(u) =g0 and R (e1 u) (13)
It is ******(ing). The specific structure of a renovation sign document makes it possible to obtain direct E (u) and R (e1 u) depending on the location of the pulse beyond one or it in the word by using the pulse response of the filter STW1 of the filter STW2 determined before equal to one. actually -- E (u) -- a degree type, i.e., :, -- [Equation 20]
E(u) =sigma [Q(n-n1)**Q (n-n2)] 2 = sigmaQ2+(n-n1) sigmaQ2**(n-n2)2sigmaQ(n-n1) - Q (n-n2)
It is if it simplifies although it becomes.;
[Equation 21]
E (u) =Eq(n1) +Eq(n2) **rho (n1 and n2) (14)
In a next door and an upper type, Eq is the energy (that is, calculated to many samples determined by the location of n1 and n2) of the suitable slanting truncated signal Q. Furthermore, R (e1 u) is [Equation 22].
R(e1 u) =R [e1 q (n1)] **R [e1 q (n2)] (15)
It is written by carrying out, is here, and is [Equation 23].

$$R\,[\,e_1\ \ q\ \ (K)\,]=\sum_{n=0}^{Ls-1-K} e_1(n+K)\ Q\ (n)$$

It comes out. A formula (14) and (15) are expressed to a single pulse word in the gestalt E(u) =Eq (n1) and R(e1 u) =R [e1 q (n1)] so that clearly from here. In order to determine the optimal excitation, the actuation performed by EM by each reverse frame is divided into three steps.
[0026] a) It is a value ai before investigating the effect of each renovation word. As soon as it can use, EM calculates and memorizes the possible value of three addends in a formula (14). As stated above, at the following reverse frame, it is a filter factor ai. Since it does not change, the count will be

performed only to the first four reverse frames. Term Eq -- a degree type, i.e., :, -- [Equation 24]

Eq(Ls-1-n)=Eq(Ls-n)+Q2 (n) (16)

(However, n= 1 .... Ls-1, Eq(Ls-1) =1)

It is alike and it is calculated that it is also by the easy repeatability procedure which followed. furthermore, a sign document -- a pair of possible value n1 of nickel2, and n2 since it contains -- count of rho -- a degree type, i.e., :, -- [Equation 25]

rhok=2Q [n2 (p)]

rhok=2rhok+1+2Q[n+n2 (p)]-Q (n)

It is alike, it follows and performs only to such a pair, and it is an upper type, has n2 (the semantics which already quoted p), i.e., n=1 ....Ls-1-n2(p), and is k=nickel2.. 1 is a value n1 and n2. It is a general pair.

b) e1 As soon as it can use an optimum value, in advance of the search procedure, EM always calculates and memorizes a value R (e1 q).

c) After such actuation and EM calculate one value of E (u) and R (e1 u) at a time, and are a value go. The word index and its related value of g which determined the related value deltaE and made the energy min are memorized.

[0027] As stated above, the trial by it being also in the word of the 1st section of a sign document, if it becomes by which a sound is not uttered is performed only when concentration of the powerful energy in the short time amount which shows, initiation, i.e., the standup, of the utterance signal section, is accepted. For this purpose, only one sample is calculated at once by shifting the aperture which chooses that group until the energy of the sample of a certain group of the corrected residual signal leaves from the start of that reverse frame within that reverse frame and all reverse frames are scanned (for example, five samples), and it memorizes which group shows the greatest energy. Furthermore, the mean power (that is, energy divided by the number of samples) in the aperture which produced maximum, and the mean power in a reverse frame are also calculated. The trial by it being also in a single pulse word is attained at them, when the ratio between the mean power in the aperture and the mean power in a reverse frame is larger than a suitable threshold in the energy of a reverse frame, and a list. Furthermore, if the optimal renovation consists of the single pulse word, the absolute value of Gain g is maximum |g| max x=|rs |max. It is restricted, is a parameter equal to about 1 here, and is |rs |max. [External Character 11]

$\widehat{h}_{min}$, $\widehat{h}_{max}$

It is the residual maximum to determine and which is calculated working. The purpose of this limit is preventing invasion to that signal of the pulse which has too high energy about the greatest residual amplitude in that same reverse frame. The early condition in Filter LTSa, STW1a, and STW2a will be updated in the end of each reverse frame. LTSa, i.e., ss, In order to update (n), it is required to add one or one pair of pulses (for it to correspond to the optimal renovation word) to ss 1 (n). yw In order to update (n), it is required to be appropriately shifted, in order to supply the value of yw2 corresponding to the optimal excitation, and to add one or two pulse responses (for it to

correspond to signal u (n)) by which Gain g was multiplied to yw1 (n). The pulse response is used also in order to update STW2a. Furthermore, since Filter STW has Degree P, only P sample (from Ls up to Ls-P) of the last of this response is important. Actuation of EM is included also in the appendix. Now, the configuration of a decoder will be described by here with reference to drawing 3 the block corresponding to what was already described with reference to drawing 1 is indicated to be by giving a figure 2 to the same reference mark. The signal with which various kinds were reconfigurated is also shown that the same reference mark used to the signal of the origin in an encoder is also.

[0028] From an encoder, a decoder lets Rhine 2a-2e pass, and receives the notation sigma to Indexes j (phi) and j (d), j (b), j, j (gmax) and j (gnor), and renovation gain (s). In each reverse frame, index j (s) directs the reverse frame which chooses renovation word s (n) in the renovation document IC 2, or does not give a renovation contribution (g= 0). Supposing a word is chosen, in M2, the absolute value will be chosen by index j(g) =j(gmax)-j (gnor) in the sign document IG 2, the gain g with the notation sigma will be multiplied, and this will give it the reconfigurated excitation signal (that is, sign document contribution of immobilization) (n) s1. This signal is the reconfigurated short-term residual signal ss. In order to give (n), it **** in the long-term composition filter LTS2. In order to make it operate completely like the replica LTS1 in an encoder, a filter LTS2 must receive the flag F which indicates the needs of performing interpolation of d and b to be Parameters d and b from the reconstruction circuit LTR 2. So, LTR2 memorizes the value of d relevant to two continuing frames, and b. and to decide whether to be the thing which is the need [to be interpolated about d and b] The read only memory which has two tables addressed by index j (d) and j (b) like LTR1 (drawing 1) out of the circuit suitable for performing the comparison described about the encoder is included. Signal ss left from LTS2 (n) is the multiplier ai which leaves Index j (phi) and is generated in the multiplier reconstruction circuit STR 2. It uses and **** in the short-term composition filter STS 2. Moreover, in STS2, the interpolated multiplier is used to the reverse frame of the beginning of each frame. reconfigurated sound signal y (n) -- still more -- linearity preliminary announcement multiplier ai from -- the multiplier obtained is used and it **** further in the adaptability filter PF inserted in the sound signal which had distortion which improves the consciousness effectiveness reconfigurated. Reconstruction signal yp ****(ed) by the output of Filter PF (n) is taken out. Probably, much more explanation will be unnecessary, since it is common knowledge for an engineer to adopt a filter like PF when encoding a sound signal. A decoder should lay on heart not taking into consideration the shift performed with the encoder. The purpose of a shift is bringing the compounded signal close to the replica of the original signal as much as possible, therefore a decoder actually needs only excitation and the information relevant to a filter.

[0029] The above publication is performed based on the un-restrictive example, and it is clear that it can accomplish without many modification and corrections deviating from the range of this invention. For example, although the sample whose amplitude is 1 is described when talking about renovation, it is also possible to use the sample as which the amplitude was chosen in the value (for example, **1, **root2, **1/root2) in the set of finite, and the signal encoded in this case includes the

information about the relative amplitude of a renovation sample. It is easy to generalize a formula (14) and (15) to the case of the pulse the amplitude of whose is not 1. Since the relative amplitude of the sample itself is quantized, selection of the sample amplitude in the value of a limited set is not restrictive. Although the timing signal over various blocks is not shown for the simplification of a drawing, the timing sequence of operation is clear from a publication there.

[0030] ** It is written that it is also by notation which is [in / the mode from which the amount in a program notation differs a little from the above-mentioned publication for the formal requirements for ****** C] different. Since the difference about existence of an inferior letter, a parenthesis, etc. does not have not clear danger, it is not discussed in a detail. n, h, rs, Eq and Q, Relq, Nip, n2 (p), [in / in notation n_ in a notation, h_ rs_ Eh, h_, Re1h Nik, n2key, and id, ib and is / a specification] Supported j (d), j (b), and j (s), and the alphabetic character "thr" added to a certain notation (Ers, Erf ....) was discussed by the specification. The threshold over each amount is shown, EPSILON and RO are the ****** factors discussed also on the specifications, and DELTA shows (DELTAd supports d' in a specification)., the increment, i.e., the increment, in a value of each amount

1) Long-term analysis /* Search to long-term preliminary announcement machine delay: */ Rrfdmax =-DBL_MAX;
As opposed to (d_=Ls;d_<=D;d_++) { Rrf[d_]=0.;
(n=K; n<=Lf+H+K-1-d_; As opposed to n++) Rrf[d_]+= rf[n+d_] * rf [n];
If it becomes (Rrf[d_] >Rrfdmax) { d[0]=d_;
Rrfdmax =Rrf [d_];
}
}

[0031]
/* 2nd order-search to the long-term preliminary announcement machine delay in the circumference of a front value: */ dmin =sround (1. -EPSILON dthr) (*d [-1]);
dmax =sround (1. +EPSILON dthr) (*d [-1]) : If it becomes (dmin <Ls) dmin =Ls;
If it becomes (dmax >D) dmax =D;
If it becomes (utterance [-1] && interpolation [-1] && (d[0] < dmin | | d [0]> dmax)) { Prfd max_=-DBL_MAX;
(d_=dmin ; d_<=dmax ; As opposed to d_++) If it becomes (Rrf[d_]>Rrfd max_) { d_=d_;
Rrfdmax_=Rrf [d_];
}
If it becomes (Rrfdmax_/Rrfdmax >=RORrf thr) d[0]=d_;
}
/* Count of a long-term preliminary announcement machine multiplier and gain; */
Erf=Erf_=Erf_[0]=0.;
if (if [ ] it 1+d[K-] [0] <=Lf+H+K-1-d [0] becomes -- {-- (n=K; n<=K-1+d[0]; n++) -- receiving -- Erf+=rf[n] * rf[n];)
(; n<Lf+H+K-1-d [0]; n++) -- receiving -- Erf_+=rf[n] * rf[n];

(; n<=Lf+H+K -1; n++) -- receiving -- Erf_[0]+=rf[n] * rf[n];

Erf+=Erf_;

Erf_+=Erf_ [0];

}

[0032]

To others As opposed to { (n=K; n<=Lf+H+K-1-d[0]; n++) Erf+=rf[n] * rf [n];

(; n<=K-1+d [0]; n++) -- receiving -- Erf_[0]+=rf[n] * rf[n];

(; n<=Lf+H+K -1; n++) -- receiving -- Erf_+=rf[n] * rf[n];

Erf_[0]+=Erf_;

}

Erf_[0]+=rf[Lf+H+K-1-d[0]] * rf [Lf+H+ K-1-d [0]];

b[0]=(Erf>=Erfthr) ?Rrf[d[0]]/Erf:0.;

G=(Erf>=Erfthr &&Erf_>=Erfthr _)?1./(1.

- b[0] * Rrf[d[0]]/Erf_:1.;

/* Correction of a long-term preliminary announcement machine multiplier: */ b max and 1

=(Erf_[-1] >=Erfthr_) ?pow (Erf_ [0]/Erf_ [-1], d(duplex) [0]/(2* Lf)) : DBL_MAX;

If it becomes (b[0] >b max1) b[0] =b max1;

If it becomes (b[0] >b max2&&G<G thr) b[0} =b max2;

/* Zero clipping of a long-term preliminary announcement machine multiplier and quantization with maintenance of a zero value: */ If it becomes (b[0]>=bq[1]/2.) {[0033]

ib=sq (N1c, bd, b [0]);

b[0]=bq[ib]

}

To others { b[0]=0.;

ib=1;

}

/* Interpolation and decision of utterance: */ if -- (duplex) (abs (d[0]-d [-1]) -- /-- d -- [- one --] -- < -- =

-- -- EPSILONdthr -- & -- & -- b -- [-- zero --] -- > -- zero . -- & -- & -- b -- [- one --] -- > -- zero .)

When it carries out, it is. Interpolation [0] =1;

To others Interpolation [0] =0;

If it becomes (G>=Gthr &&b[0] >=bthr) Utterance [0] =1;

To others Utterance [0] =0;

/* Alternative correction of long-term preliminary announcement machine delay: */ If it becomes (interpolation [0] &&! utterance [0]) { DELTAd=sround (/(GAMMA* Lf)) (duplex) (h_* d [0]);

If it becomes (DELTAd<-absDELTAdmax) DELTAd=-absDELTAdmax ;

If it becomes others (DELTAd>absDELTAdmax) DELTAd=absDELTAdmax ;

If (d[0]+DELTAd>=Ls&&d[0]+DELTAd< =D&& (duplex) abs (d[0]+DELTAd-d[-1] / d[-1]

<=EPSILONdthr) if it becomes d[0]+=DELTAd;)

}

/* Long-term preliminary announcement machine delay or display of a zero multiplier: */ If (b [0]>0.] if it becomes id=d[0]- Ls+1;)

To others id=N1d;

[0034] 2) In this part of the minimization notation of error energy, "bang" shows recognition of a standup.

/* Preparation to a time amount shift and a renovation search: */ If (interpolation (0] | | utterance [0]) if it becomes { Ers=Ers_=0.;)

(n=o+Ls-d[0]+h; As opposed to n<=o+h-1;n++) Ers+=rs[n] * rs [n];

(; n<=o+Ls+h -1; n++) -- receiving -- Ers_+=rs[n] * rs[n];

absrs max=-DBL_MAX;

(n=GAMMA* o+h_;n<=GAMMA*+(o+Ls) h_-1 As opposed to ;n++) { absrs=fabs (rs_ [n]);

If it becomes (absrs>absrs max) {[0035]

np[0]=n;

absrs max=absrs;

}

}

np[0]-=GAMMA* o;

If If it becomes { hmin =-GAMMA* H; (Ers_>=Ersthr &&d[0] * Ers_/ (Ls*) >(Ers+Ers_) =ROPrsthr)

If it becomes (peak &&np[-1] <=h_-1&&hmin < np[-1]) hmin =np [-1];

hmax =GAMMA* H;

If it becomes (hmax > np [0]) hmax =np [0];

DELTAhpL=h_-hmin ;

DELTAhpH=hmax -h_;

If (DELTAhpL>=sround (GAMMA* DELTAhp))

& If it becomes &DELTAhpH>=sround (GAMMA* DELTAhp) { hmin =h_-sround (GAMMA* DELTAhp);

hmax =h_+sround(GAMMA* DELTAhp);

}

To others If it becomes (DELTAhpL>=DELTAhpH) hmin = max (hmax-2* sround (GAMMA* DELTAhp) and hmin);

To others hmax = min (hmin+2* sround (GAMMA* DELTAhp) and hmax);

Peak = 1;

}

To others { hmin =hmax =h_;

Peak = 0;

}

}

[0036]

others {-- -- if it becomes (h_<0) -- hmin =hmax = min(h_+sround (GAMMA* DELTAhr), 0);

To others hmin =hmax = max (h_-sround (GAMMA* DELTAhr), 0);
Peak = 0 }
If it becomes (1<=Nis_-1) { h_[0] =Eh[Ls-1] =1.;
(n=1,n_=Ls-2; n<=Ls-1; n++,n_--)
It is alike and receives. { h_[n]=0.;
(k=1; k<=min (n, P); As opposed to k++) h_[n]+=aw[l] [k] * h_ [n-k];
Eh[n_] =Eh[n_+1]+h_[n] * h_ [n];
}
(-- i=Nik and j=nickel2; i>=1; i --) -- receiving -- { ro[j] =2.* h_ -- [-- n2 key [i]];
(n= 1, j --; n<=Ls-1-n2 key [i];)
n++ and j -- receiving -- ro[j]= -- ro[j+1]+2.* h_[n+n2 key [i]]
* h_ [n];
}
}
/* Search to the time amount shift of a short-term preliminary announcement residual signal: */
(n=o and n<=o+Ls-1; n++) It receives. {[0037]
xw [n] =0.;
(k=1; k<=P; As opposed to k++) xw [n]+=aw[l] [k] * xw [n-k];
e1 [n-o] =xw [n]-yw [n];
}
Eelmin =DBL_MAX;
(i=0,h_=hmax ; i<=GAMMA-1&&h_>=hmin ;
i++ and h_ -- receiving -- { Ee1=0.;
(n=0,n_=GAMMA* o+h_; n<=Ls-1;
As opposed to n++ and n_+=GAMMA { xw 2[i] [n] =rs_ [n_];
(k=1; k<=min (n, P); As opposed to k++) xw 2[i] [n]+=aw[l] [k] * xw 2 [[i] n-k];
e1_=e1[n]+xw 2 [[i] n];
Ee1+=el_* el_;
}
If it becomes (Ee1<Eelmin) { h_=h_;
Ee1min =Ee1;
}
}
(i=0,n_=GAMMA* o+h_; h_>=hmin ;
i=(i<GAMMA -1) ?i+1:0 and h_ -- and n_ -- receiving -- { Ee1=0.;
(-- n=Ls -1; n>=1; n --) -- receiving -- {[0038]
xw 2[i] [n] =xw 2[i] [n-_[1]+h] [n] * rs_ [n_];
e1_=e1[n]+xw 2 [[i] n];
Ee1+=e1_* e1_;

}
xw 2[i][0]=rs_ [n_];
e1_=e1[0]+xw 2 [[i] 0];
Ee1+=e1_* el_;
If it becomes (Eel<Ee1min) { h_=h_;
Ee1min =Ee1;
}
}
h=sround (duplex) (h_/GAMMA);
/* count of the long-term component of the error by which weighting was carried out to the voice list by which weighting was carried out -- Preparation:*/to a ****** search (n=o, n_=GAMMA* o+h_; n<=o+Ls-1;)
n++ As opposed to n_+=GAMMA { xw [n] =rs_ [n];
(k=1; k<=P; As opposed to k++) xw [n]+=aw[l] [k] * xw [n-k];
e1 [n-o] =xw [n]-yw [n];
sqrs[n-o] =rs_[n_] * rs_ [n_];
}
(k=o; k<=Ls-1; As opposed to k++) { Re1h[k] =e1 [k];
(n=1; n<=Ls-1-k; As opposed to n++) Re1h[k]+=e1 [n+k] * h_ [n];
}
[0039]
/* Alternative renovation gain clipping and preparation to an alternative renovation search: */ If it becomes (utterance [0]) absg1 max=nu* absrs max;
To others { Ers=0.;
(n=0Ers+=sqrs; n<=B-1 (n];); As opposed to n++)
Ers max=Ers_=Ers;
(; n<=Ls-1; n++) -- receiving -- { Ers+=sqrs[n];
Ers_+=sqrs[n]-sqrs [n-B];
If it becomes (Ers_>Ers max) Ers max=Ers_;
}
If it becomes (Ers>=Ers thr_&&Ls* Ers max/ (B* Ers) >=ROPrs thr_) bang=1;
To others bang=0;
}
[0040]
/* Alternative search to the renovation parameter accompanied by alternative renovation gain clipping : */ n1[l] =n2[l] =0;
g1[l]=g2[l]=0.;
DELTAEe1e max=-DBL_MAX;
If it becomes (utterance [0] | | bang) (n1_=0, i=1;n1_<=Ls-1;n1++, i++)

It is alike and receives. { Eu=Eh [n1_];
Re1 u=Re1h [n1_];
g1_=Re1u/Eu;
DELTAEele=g1_* Re1u;
If it becomes (DELTAEele>DELTAEele max) { n1[l]=n1_;
is[l]=i;
g1[l]=g1_;
DELTAEele max=DELTAEele;
}
}
To others i=Ls +1;
If (g1[l] <-absg1 max) If it becomes g1[l] =-absg1 max; (utterance [0])
If it becomes others (g1[1] >absg1 max) g1[l] =absg1 max;
(j=1, k= 1 (n1_=0, n2_=n2 [key j];n2_<=Ls-1;); j<=Nik; As opposed to j++)
n1_++ n2_++ i++ As opposed to k++ { Eu=Eh[n1_]+Eh[n2_]-ro [k];
Re1 u=Re1h[n1_]-Re1h [n2_];
g1_=Re1u/Eu;
DELTAEele=g1_* Re1u;
[0041]
If it becomes (DELTAEele>DELTAEele max) { n1[l]=n1_;
n2[l]=n2_;
is[l]=i;
g1[l]=g1_;
g2[l] =-g1_;
DELTAEele max=DELTAEele;
}
i++;
Eu=Eh[n1_]+Eh[n2_]+ro [k];
Relu=Re1h[n1_]+Re1h [n2_];
g1_=Relu/Eu;
DELTAEele=g1_* Relu;
If it becomes (DELTAEele>DELTAEele max) { n1[l]=n1_;
n2[l]=n2_;
is[l]=i;
g1[l]=g1_;
g2[l]=g1_;
DELTAEele max=DELTAEele;
}

}

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]

[Drawing 1] The block diagram of an encoder

[Drawing 2] The functional diagram of the block with an encoder

[Drawing 3] The block diagram of a decoder